

Equalization of Data Channels
with Feedback Filters

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Papers Included in the Dissertation

- I An Equalizer with Feedback Filter.
Ericsson Technics 28 (1972):2 pp 57 - 101

- II On the Performance of Time Discrete Feedback Filters as
Equalizers for Data Channels.
The Lund Institute of Technology, Telecommunication Theory
Technical Report TR-41 1973

- III Realization of an Adaptive Receiver for Data Transmission.
The Lund Institute of Technology, Telecommunication Theory
Technical Report TR-42 1973 (in Swedish co-author Tommy Svensson)

- IV Linear Network Synthesis with Laguerre Polynomials.
Ericsson Technics 27 (1971):2 pp 83 - 109

Introduction

The existing telephone network allows a communication channel to be established between practically any two places. When the need for data transmission arose it was natural to use the telephone network as transmission medium. It proved to be an excellent means of conveying data at low data rates.

The use of digital computers has introduced an increasing need for fast communication of data. The noise margin on the telephone network is also sufficient to permit a very high data rate. However, by increasing the data rate the linear distortion of the channel often becomes a limiting factor. The linear distortion causes intersymbol interference, i.e., the distortion of pulses by the channel results in these pulses being smeared out in time so as to overlap other transmitted pulses. To alleviate the effects of intersymbol interference it is necessary to equalize the channel. In practice the channel is often random in the sense of being one of an ensemble of possible channels. At high-speed data transmission the equalizer has to be automatically adjustable. Various methods of achieving automatic equalization have been proposed in recent years [1] - [7]. Equalization of a channel can be arranged in the following way. When the connection has been completed, a known signal pattern is to be sent. The output signal from the equalizer is compared with the desired output signal. The error, i.e., the difference between these two signals, is used to control the parameters of the equalizer. These are adjusted to minimize a given error criterion, e.g. the square error or the peak distortion, for a particular channel. After this training period the transmission of information can begin.

We want to make the equalizer as effective as possible for each chosen channel with a minimum number of adjustable parameters. As measures of the effectiveness we use the residual distortion and the adjustment time, i.e., the length of the training period. The residual distortion is defined as the distortion in the output signal with the parameters optimally adjusted.

The present studies deal with the realization of an equalizer that meets given performance criteria with a minimum number of adjustable parameters. In judging a realization we use the residual distortion and the adjustment time.

Model of the Data Communication System

We consider a synchronous, coherent, linearly modulated data communication system [9]. In each time interval of length T one data symbol is transmitted. On the channel we have a pulse-amplitude-modulated (PAM) signal.

The channel is assumed to be linear. It is selected at random with given probability from a finite ensemble of channels. The channel is characterized by its frequency function or by its impulse response. We assume that the channel is noiseless.

The receiver consists of a linear network i.e. the equalizer, followed by a threshold detector. The output signal of the equalizer is sampled at $t=kT$. The sample values are compared with given levels in the threshold detector and an estimate of the transmitted data symbol is made. We are thus interested in the output signal of the equalizer only at equidistant time points.

Adjustment Algorithms

One realization of automatically adjustable equalizer is the transversal filter. By adjusting the tap gains the output signal is forced to approximate the desired output signal. The error, i.e. the difference between these two signals, is a linear function of the tap gains. This means that the considered error criterion e.g. the square error or the peak distortion, is a convex function of the tap gains. Thus a unique optimum exist. [10]

For either criterion the optimal tap gains are obtained as the solution of a linear equation system. [1]. However, implementation in real time of this solution will be extremely complicated. From a practical point of view a better method would be to find the minimum in an iterative way.

For minimizing the peak distortion an iterative scheme for solution of the corresponding linear equation system, which is particularly easy to implement, was discovered by Lucky [2]. The tap gains were preset by sending isolated data pulses before transmitting information. Later Lucky [3] modified the equalizer to work adaptively, i.e. the equalizer information is derived directly from the received data signals. In this way changes in the characteristics of the channel could be taken into account. The two papers [2] and [3] give an exhaustive analysis of the transversal filter equalizer.

Later on, other iterative algorithms for solution of the linear equation system, some of which can be found in the mathematical literatures [8], have been suggested and implemented. Some of these minimize the peak distortion [4], others minimize the square error [5].

However, it might be interesting to investigate other structures of the equalizer than the transversal filter. A slight generalization is the feedback transversal filter or the recursive filter, which contains the transversal filter as a special case namely when all the feedback coefficients are zero.

Under certain conditions, viz. a time-limited input signal or an input signal, the Z-transform of which is a rational function, a recursive filter eliminates all lagging echoes. Thus, in this way, a finite recursive filter is equivalent to a transversal filter with an infinite number of delay cells.

Paper I contains an investigation of methods of adjusting the feedforward as well as the feedback parameters of the recursive filter. Furthermore, methods of determining the instant, when the main pulse shall appear, are described. The analysis is further extended in Paper III, which also describes the realization of and measurements on an adaptively adjustable recursive filter equalizer. In this paper we also study some aspects of the performance of the feedback equalizer in noise.

Equalization of an Ensemble of Channels.

In the above mentioned realizations of the equalizer all parameters are automatically adjusted to fit a particular channel. However, by using the knowledge about the a priori channel statistics some parameters of the equalizer can be calculated to fit the given ensemble of channels. The number of adjustable parameters required to fulfil given performance criteria is thus smaller than if all parameters are adjustable.

Maurer and Franks [6] were among the first to use a priori known channel statistics to determine parameters common for all channels. They used a parallel-path configuration with adjustable path gain as equalizer structure. The transfer function for each path were determined to minimize the square error averaged over the channel statistics. When the ensemble consists of all-pass channels with random phase functions the exact solution is given as the solution of an eigenvalue problem.

A similar structure as in [6] has been used by Chang [7], who considered the design of rapid start-up equalizers. However, no averaging over a priori known channel statistics is performed. In Paper II we assume that the ensemble consists of a finite number of linear independent channels. Each channel appears with a given probability. We use the same general structure as in [6]. However, we demand, that the path transfer function be rational functions, the order of which are at most equal to the number of paths. To decrease the adjustment time we modify the structure by a method similar to the one described by Chang [7]. However, our method is more flexible and more easily calculated. After this modification the adjustment time will be approximately independent of the denominator of the path transfer function, if the distortion of the output signal can be made small. Thus the denominators are determined once for all so that the square error averaged over the ensemble of channels is minimized. The calculation is made iteratively by using the algorithm, which is derived in Paper I.

Relations to the Approximation Problem

The problem of equalizing a data channel is closely related to the problem of synthesising a system with given in- and output signals.

Paper IV contains an investigation of the performance of a time-continuous feedback equalizer, where the feedforward as well as the feedback parameters can be automatically adjusted. The first part of Paper IV describes a method of representing a time-continuous impulse response by a polynomial using Laguerre functions. This enables methods from the theory of time-discrete systems to be used to determine a rational system function, the impulse response of which is approximately equal to the desired impulse response.

The second part of Paper IV deals with the problem of determining the parameters of a given network so that a given input signal gives rise to a given output signal. By looking at the output signal only at equidistant points the input-output relation can be described by a difference equation. Then the parameters of the system can be automatically adjusted with essentially the same iterative algorithm as is described in Paper I.

References

1. Lucky R.W., Salz J., Weldon E.J.
"Principles of Data Communication"
McGraw-Hill N.Y. 1968
2. Lucky R.W.
"Automatic equalization for digital communication"
BSTJ vol 44, April 1965, pp 547 - 588
3. Lucky R.W.
"An Automatic Equalizer for General Purpose Communication Channels"
BSTJ vol 45, 1966, pp 255 - 286
4. Lytle D.W.
"Convergence Criteria for Transversal Equalizers"
BSTJ vol 47, No 8, 1968, pp 1775 - 1800
5. Gersho A.
"Adaptive Equalization of Highly Dispersive Channels for Data Transmission"
BSTJ vol 48, 1969, pp 55 - 70
6. Maurer R.E., Franks L.E.
"Optimal Linear Processing of Randomly Distorted Signals"
IEEE Trans. on Circuit Theory, CT-17, No 1, 1970, pp 61 - 67
7. Chang R.W.
"A New Equalizer Structure for Fast Start-Up Digital Communications"
BSTJ vol 50, 1971, pp 1969 - 2014
8. Varga R.S.
"Matrix Iterative Analysis"
Prentice-Hall 1962
9. Bennet W.R., Davey J.R.
"Data Transmission"
McGraw-Hill 1965

10. Rice J.R.

"The Approximation of Functions" Vol 1.

Addison-Wesley Publishing Company, 1964

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